Lab Manual 01

The effects of Sampling in Discrete Time Signals

Lab Objectives:

- To study the relationship between discrete-time and continuous time signals by examining sampling and aliasing.

Learning Outcomes:

Upon completing this lab, students will be able to:

- Define Sampling
- Explain Discrete-Time Signals
- Understand Sampling Rate
- Relate Sampling to Discrete-Time Signals
- Explore Aliasing
- Identify Aliased Signals

Signals:

Signals are physical quantities that carry information in their patterns of variation. Signal can be a function of time, distance, position, temperature, and pressure etc. for example, the traffic signals which changes with the time. So, the traffic signal is a function of time.

Continuous time vs Discrete time signals

Continuous time signals are continuous functions of time denoted by x(t), while discrete-time signals are sequences of numbers, denoted by x[n]. If the values of a sequence are chosen from a finite set of numbers, the sequence is known as a digital signal.

Continuous-time, continuous-amplitude signals are also known as analog signals. Analog phenomenon is continuous – like a human speech of speaker, or a continuously rotating disc attached to the shaft of motor etc. With analog phenomena, there is no clear separation between one point and the next; in fact, between any two points, an infinite number of other points exist.

Discrete phenomenon, on the other hand, is clearly separated. There's a point (in time or space), and then there's a neighboring point, and there's nothing between the two.

Signal Processing

Signal processing means analyzing the signal and performing operation on it. It is concerned with the acquisition, representation, manipulation, transformation, and extraction of information from signals.

In analog signal processing, these operations are implemented using analog electronic circuits. Converting the continuous phenomena of images, sound, and motion into a discrete representation that can be handled by a computer is called analog-to-digital conversion.

Digital signal processing

Digital signal processing involves the conversion of analog signals into digital, processing the obtained sequence of finite precision numbers using a digital signal processor or general purpose computer, and, if necessary, converting the resulting sequence back into analog form. When stored in a digital computer, the numbers are held in memory locations, so they would be indexed by memory address.

Regardless of the medium (either sound or an image), analog-to-digital conversion requires the same two steps:

Sampling: It is phenomena of converting CTS to DTS.

This operation chooses discrete (finite) points at which to measure a continuous phenomenon (which we will also call a signal). In the case of sound, the sample points are evenly separated in time. In the case of images, the sample points are evenly separated in space.

Sampling Rate: The number of samples taken per unit time or unit space is called the sampling rate. The frequency of sampled/discrete phenomenon (signal) can be calculated as:

fd = F /Fs (cycles/sec)/(samples/sec) = cycles/ samples

Where, F = Frequency of analog or continuous phenomenon (signal). [Unit: cycles/sec]

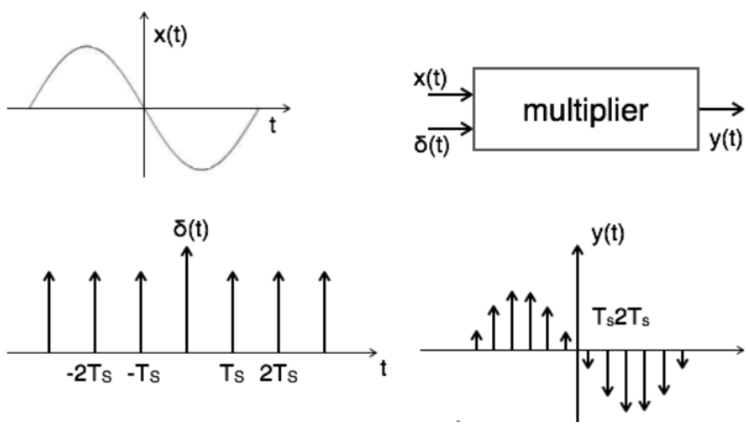
Fs = Sampling frequency or sampling rate [Unit: samples/sec]

fd = Normalized Frequency of Discrete phenomenon (signal). [Unit: cycles/sample]

Sampling Theorem:

A continuous time signal can be represented in its samples and can be recovered back when sampling frequency fs is greater than or equal to the twice the highest frequency component of message signal. i. e.

$$fs \geq 2fm.$$



Aliasing:

A common problem that arises when sampling a continuous signal is aliasing, where a sampled signal has replications of its sinusoidal components which can interfere with other components. It is an effect that causes two discrete time signals to become indistinct due to improper sampling. If the frequency of the continuous signal (F0) is greater than half of the sampling frequency (Fs/2), aliasing may occur or we can say that when sampling frequency is less than twice of input frequency signal, then it causes an aliasing effect.

$$Fs<2fm \text{ or } F0<Fs/2$$

PROCEDURE:

1. Simulate and plot two CT signals of 10 Hz and 110 Hz for 0 < t < 0.2 secs.
2. Sample at Fs = 100 Hz and plot them in discrete form.
3. Observe and note the aliasing effects.
4. Explore and learn.

STEPS:

1. Make a folder at desktop and name it as your current directory within MATLAB.
2. Open M-file editor and type the following code:

```
clear all;
```

```
close all;
clc;
F1 = 10;
F2 = 110;
Fs = 100;
Ts = 1/Fs;
 t = [0 : 0.0005 : 0.2];
x1t = cos(2*pi*F1*t);
x2t = cos(2*pi*F2*t);
figure, plot(t,x1t,t,x2t, 'LineWidth',2);
xlabel('cont time (sec)'); ylabel('Amp');
xlim([0 0.1]); grid on; legend('10Hz','110Hz');
title('Two CTCV sinusoids plotted');
```

3. Save the file as P011.m in your current directory and 'run' it, either using F5 key or writing the file name at the command window. (Check for the correctness of the time periods of both sinusoids.)
4. Now add the following bit of code at the bottom of your P011.m file and save.

```
nTs = [0 :Ts : 0.2]; %This line creates a vector nTs with values ranging from 0 to 0.2 with a step size of
Ts. The variable Ts represents the sampling interval.
n = [1 : length(nTs)-1 ]; %This line creates a vector n representing the discrete time indices. It starts
from 1 and goes up to the length of nTs minus 1.
x1n = cos(2*pi*F1*nTs); % This line generates a discrete-time signal x1n by sampling a cosine wave
with a frequency of F1 at the time instances given by nTs
x2n = cos(2*pi*F2*nTs);
figure, subplot(2,1,1), stem(nTs,x1n,'LineWidth',2);
grid on;
xlabel('discrete time (sec)');
ylabel('Amp'); xlim([0 0.1]);
subplot(2,1,2),
stem(nTs,x2n,'LineWidth',2);
grid on; title('110Hz sampled')
xlabel('discrete time(sec)');
 ylabel('Amp'); xlim([0 0.1]);
```

5. Before hitting the 'run', just try to understand what the code is doing and try to link it with what we have studied regarding concepts of frequency for DT signals.
6. Now 'run' the file and observe both plots. To see what is really happening, type the following code at the bottom of your existing P011.m file and run again.

```
figure, plot(t,x1t,t,x2t);
hold; stem(nTs,x1n,'r','LineWidth',2);
xlabel('time (sec)'); ylabel('Amp');
xlim([0 0.05]); legend('10Hz','110Hz');
```

7. Observe the plots.

RESULT:

Observe the cause and effects of what you just saw.

LAB TASKS:

1. Consider the following CT signal:

x(t) = sin (2 pi F0 t).

The sampled version will be x(n) = sin (2 pi F0/Fs n),

where n is a set of integers and sampling interval Ts=1/Fs.

Plot the signal x(n) for n = 0 to 99 for Fs = 5 kHz and F1 = 0.5, 2, 3 and 4.5 kHz.

Explain the similarities and differences among various plots.

2. Generate a tone in MATLAB with varying frequency:

f = 1000,2000,3000,4000, 5000, 6000, 8000, 9000, 25000, -1000, -2000, -3000 Hz with Fs = 8000 samples/sec. Listen to the tones, and observe at Sounds like what frequency? Also Specify whether Aliasing is happening or not.

Use the function 'sound' to generate tones of different frequencies.

Add a small pause between tones.